

# Architecture V1.1.1 (2003-10)

---

*Technical Specification*

## **Push-to-talk over Cellular (PoC); Architecture; PoC Release 1.0**

---

Ericsson, Motorola, Siemens, Nokia

**Keywords**

---

Push-to-talk over Cellular (PoC), Architecture

---

**Copyright Notification**

No part may be reproduced except as authorized by written permission of the contributing companies.

© Ericsson, Motorola, Siemens, Nokia  
All rights reserved.

---

## Contents

<u>Introduction</u>	5
1 <u>Scope</u>	6
2 <u>References</u>	6
3 <u>Definitions and abbreviations</u>	7
3.1 <u>Definitions</u>	7
3.2 <u>Abbreviations</u>	8
3.3 <u>Requirement vocabulary</u>	9
4 <u>Architecture</u>	10
5 <u>Description of functional entities</u>	11
5.1 <u>User Equipment (UE)</u>	11
5.2 <u>IMS Core</u>	11
5.3 <u>Group and List Management Server (GLMS)</u>	11
5.4 <u>PoC Server</u>	11
5.5 <u>Presence Server</u>	12
6 <u>Description of the interfaces</u>	12
6.1 <u>Interface I<sub>s</sub>: UE – IMS Core</u>	12
6.2 <u>Interface I<sub>f</sub>: IMS Core – PoC Server</u>	12
6.3 <u>Interface I<sub>t</sub>: UE – PoC Server</u>	12
6.4 <u>Interface I<sub>m</sub>: UE – GLMS</u>	12
6.5 <u>Interface I<sub>k</sub>: PoC Server – GLMS</u>	13
6.6 <u>Interface I<sub>p</sub>: IMS core – Presence Server</u>	13
6.7 <u>Interface I<sub>pl</sub>: Presence Server – GLMS</u>	13
7 <u>System concepts</u>	13
7.1 <u>Identification</u>	13
7.1.1 <u>Address of record (a.k.a. public user identity)</u>	13
7.1.2 <u>Private user identity</u>	13
7.1.3 <u>Group identities</u>	13
7.1.4 <u>Contact list identities</u>	14
7.2 <u>Addressing</u>	14
7.2.1 <u>IP addresses</u>	14
7.2.2 <u>Phone numbers</u>	14
7.2.3 <u>SIP URI</u>	14
7.3 <u>Routing principles</u>	14
7.3.1 <u>Registration</u>	14
7.3.2 <u>Session establishment</u>	14
7.3.3 <u>User location</u>	15
7.4 <u>Security</u>	15
7.4.1 <u>Threats</u>	15
7.4.2 <u>Countermeasures</u>	15
7.5 <u>Floor control</u>	16
7.6 <u>Codecs</u>	16
7.7 <u>Signaling compression</u>	16
7.8 <u>User plane adaptation</u>	17
7.9 <u>Charging</u>	17
7.10 <u>Roaming</u>	17
7.11 <u>Presence</u>	17
7.12 <u>Do-not-Disturb</u>	17
8 <u>High level procedures</u>	17
8.1 <u>PDP Contexts</u>	18
8.2 <u>DNS name server discovery</u>	18
8.3 <u>DNS procedures</u>	18
8.3.1 <u>SIP URI resolution</u>	18

<u>8.3.2</u>	<u>HTTP URL resolution</u>	18
<u>8.4</u>	<u>Early Session dialog establishment</u>	18
<u>8.5</u>	<u>Instant Personal Talk</u>	19
<u>8.6</u>	<u>Ad-hoc Instant Group Talk</u>	20
<u>8.7</u>	<u>Instant Group Talk</u>	20
<u>8.8</u>	<u>Chat Group Talk</u>	20
<u>8.9</u>	<u>Instant Personal Alert</u>	21
<u>ANNEX A (informative): Known 3GPP/OMA divergence</u>		22
<u>Annex B: Change History (Informative)</u>		23

## Foreword

This Technical Specification has been produced by Ericsson, Siemens, Motorola and Nokia.

---

## Introduction

This technical specification describes an architecture that fulfils the user requirements described in reference [1].

The basic architecture is described in clause 4.

The functional entities of the architecture are described in clause 5.

The interfaces of the architecture are described in clause 6.

General system concepts, that have architectural implications, are described in clause 7.

An overview of the high level procedures, for informational purposes, is described in clause 8.

## 1 Scope

This document describes the functional entities, interfaces, system concepts and high-level procedures of the Push-to-Talk over Cellular (PoC) service.

The PoC service is characterized by a half duplex form of communication whereby one user will communicate with other users by pressing a button (or by performing equivalent action) on an PoC enabled User Equipment (UE).

This specification is part of PoC release 1.0.

## 2 References

The following documents contain provisions, which through reference in this text constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a PoC document, a non-specific reference implicitly refers to the latest version of that document in the same Release as the present document.

- [1] Push-to-Talk over Cellular User Requirements; PoC Release 1.0
- [2] PoC Signaling Flows; PoC Release 1.0
- [3] PoC User Plane; Transport Protocols; PoC Release 1.0
- [4] PoC User Plane; (E)GPRS Specification; PoC Release 1.0
- [5] PoC List Management and Do-Not-Disturb; PoC Release 1.0
- [6] 3GPP TS 23.228 "Technical Specification Group Services and System Aspects; IP Multimedia Subsystem (IMS); Stage 2 (Release 6)"
- [7] 3GPP TS 24.008 "Mobile radio interface Layer 3 specification; Core network protocols; Stage 3"
- [8] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on SIP and SDP; Stage 3 (Release 5)"
- [9] IETF RFC 1332 "The PPP Internet Protocol Control Protocol (IPCP)"
- [10] IETF RFC 1877 "PPP Internet Protocol Control Protocol Extensions for Name Server Addresses"
- [11] IETF RFC 1889 "RTP: A Transport Protocol for Real-Time Applications"
- [12] IETF RFC 2616: "Hypertext Transfer Protocol HTTP/1.1"
- [13] IETF RFC 3261: "SIP: Session Initiation Protocol"
- [14] IETF RFC 3263: "Session Initiation Protocol (SIP): Locating SIP Servers"
- [15] IETF RFC 3320: "Signaling Compression (SigComp)"
- [16] IETF RFC 3321: "Signaling Compression (SigComp) - Extended Operations"
- [17] IETF RFC 3485: "The Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Static Dictionary for Signaling Compression (SigComp)"
- [18] IETF RFC 3486: "Compressing the Session Initiation Protocol (SIP)"

- [19] IETF <[draft-ietf-sip-callerprefs-09.txt](#)> (expire December 29, 2003): "Caller Preferences for the Session Initiation Protocol (SIP)"
  - [20] IETF <[draft-ietf-simple-presence-10.txt](#)> (expires July 2003): "A Presence Event Package for the Session Initiation Protocol (SIP)"
  - [21] ITU-T Recommendation E.164: "The international public telecommunication numbering plan"
  - [22] IETF RFC 2486: "The Network Access identifier".
  - [23] IETF RFC 2806 "URLs for Telephone Calls"
- 

## 3 Definitions and abbreviations

### 3.1 Definitions

For the purposes of the PoC specifications, the following terms and definitions apply.

**Access control:** Each PoC user can define rules that describe who is allowed to contact him/her using the PoC service. The PoC Server implements the access control policy according to these defined rules.

**Access list:** Each PoC user has two access lists: a user accept list and user reject list. Access lists are used for controlling whether the PoC server is allowed or not to send talk session requests to the user when requested by other user.

**Ad-hoc instant group talk:** A feature providing a user to ad-hoc establishes a PoC session with other PoC users.

**Chat group:** A persistent group created for chat group talk. Each group member joins the talk session individually.

**Chat group talk:** A feature providing users with the capability to join a pre-defined chat group. The chat group may be open or restricted.

**Chat talk session:** A talk session established by a chat group talk.

**Contact list:** A list available to the end user containing the addresses of other users or groups.

**Contact:** A contact is an identity of a user, or a group. A contact includes the SIP URI or a TEL URI of the entity, type of the entity (user or group) and optionally the display name.

**Floor control:** A control mechanism that arbitrates requests, from the UEs, for the right to speak.

**Group Talk:** An instant group talk, ad-hoc instant group talk or chat group talk.

**Group:** Group is predefined set of users together with its attributes. The group is used for easy session establishment and/or for defining session access policy. Each group is identified by its SIP URI.

**Instant group:** A persistent group created for instant group talk. The users PoC server invites all the other group members to a talk session.

**Instant group talk:** A feature providing a user to establish a PoC session with other members in a pre-defined instant group. The instant group is always a restricted group.

**Instant personal Alert:** A feature providing a user with the capability to send a callback request to another user.

**Instant personal talk:** A feature to establish a PoC session with another user.

**Instant talk session:** A talk session established by instant personal talk, ad-hoc group talk or instant group talk.

**Invited user:** This is the PoC user who has been invited to a talk session.

**Inviting user:** This is the PoC user inviting other PoC user(s) to the to a talk session.

**maxptime:** The maximum amount of media which can be encapsulated in a RTP payload packet, expressed as time in milliseconds. The time is calculated as the sum of the time the media present in the packet represents. The time should be a multiple of the frame size. In PoC the allowed values are N\*20; where N>0 and N<21.

**media capabilities list:** In this list, the PoC Server shall store the downlink media capabilities of all UEs that are active in sessions served by the PoC Server.

**media capabilities:** A set of parameters that should describe the performance of the PoC user equipment (UE), the speech coder used and the performance of the radio bearer that carries the PoC service (the quality of service parameters agreed upon etc).

**media parameters:** The PoC Server uses the media capabilities list to determine the settings the user equipments should use in the talk session. The information transmitted from the PoC Server to the UE in order to alter the settings of the UE, is in this document referred to as media parameters. Media parameters are transmitted by SIP/SDP messages.

**mode-set:** Restricts the active codec mode set to a subset of all modes. Possible values are a comma separated list of modes from the set: 0,...,7. If the decoder specifies such mode set, the encoder MUST abide by the request and MUST NOT use modes outside of the subset. If not present, all codec modes are allowed for the session.

**Open group:** A group that can be joined by any user.

**Participant:** A PoC user in talk session.

**ptime:** Number of frames per RTP-packet the UE needs to be able to receive the media stream on its downlink. Ptime is given as the length of time in milliseconds represented by the media that needs to be in a RTP packet.

**Restricted group:** A group that can be joined only by predefined user(s).

**Session:** A session is considered as an exchange of data between associations of participants.

**Talk session:** This is an established connection between PoC users where the users can communicate one at a time in a half duplex manner.

**talk spurt:** A part of the speech signal that starts with a speech onset and ends when the speech coder goes down in DTX-mode. Hence a talk burst can consists of several talk spurts.

**Talk-burst:** The media recording, transport and playback that occur from when the user has the floor.

**User:** A human using the described features through a terminal device.

**User accept list:** User accept list is a list of items each identified by its SIP URI.

**User equipment:** User equipment is a hardware device (e.g. phone) with Push-to-Talk software used by users.

**User reject list:** User reject list is a list of items each identified by its SIP URI.

## 3.2 Abbreviations

For the purposes of the PoC specifications, the following abbreviations apply:

AMR	Adaptive Multi Rate
DNS	Domain Name System
DTD	Document Type Definition
FQDN	Fully Qualified Domain Name
FFS	For Further Study
GERAN	GSM/EDGE Radio Access Network
GGSN	Gateway GPRS Support Node
GLMS	Group and List Management Server
HTTP	Hypertext Transfer Protocol
IMS	IP multimedia subsystem
IPCP	Internet Protocol Control Protocol
ISC	IMS service control interface
MD5	Message Digest no. 5
N/A	Not Applicable

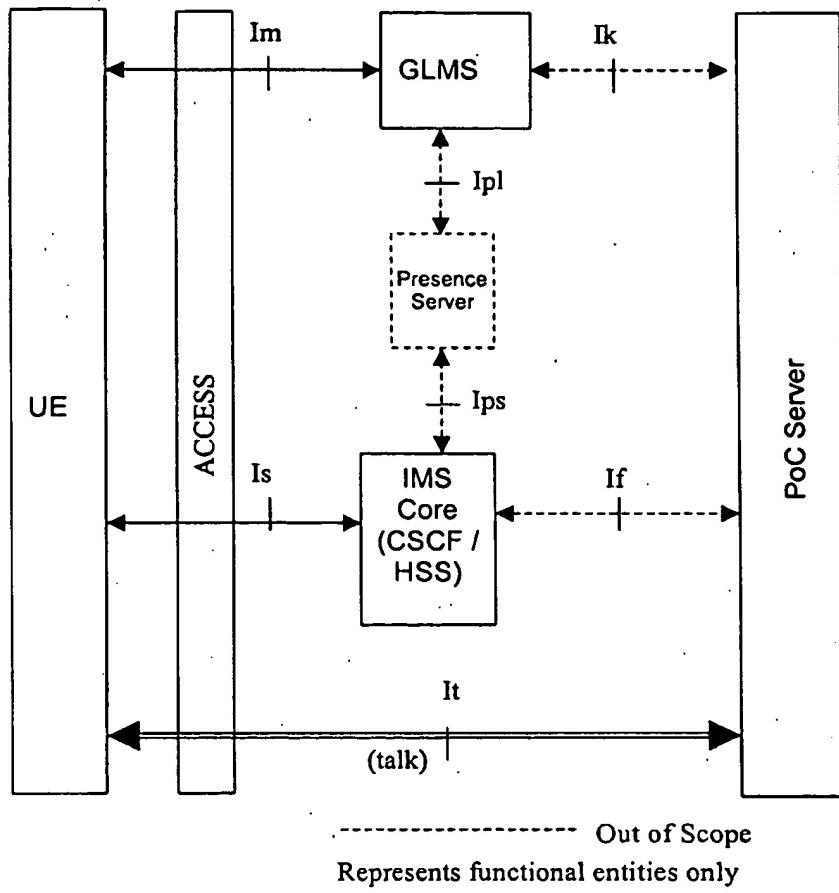
NAI	Network Access Identifier
NAPTR	Naming Authority Pointer
P-CSCF	Proxy-CSCF
PCO	Protocol Configuration Options
PDP	Packet Data Protocol
PoC	PTT over Cellular
PTT	Push to Talk
RTCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SRV	DNS resource record for the location of services
UCS	Universal Character Set
UE	User Equipment
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
UTF-8	UCS Transformation Format 8
UTRAN	Universal Terrestrial Radio Access Network.
XML	Extensible Mark-up Language

### 3.3 Requirement vocabulary

Shall	Indicates a mandatory requirement.
Should	Indicates a recommendation.
May	Indicates an optional requirement.

## 4 Architecture

The PoC functional architecture is outlined in Figure 1.



**Figure 1: PoC architecture**

**NOTE:** The access in the PoC architecture includes both the radio access as well as the other nodes required to gain IP connectivity (e.g. GSNs).

PoC shall be based on IMS as specified in 3GPP TS 23.228 [6] and TS 24.229 [7], with exceptions regarding to:

- IPv4 only is supported.
- The IMS-enabled PoC service may be offered through mobile networks different from that specified in 3GPP release 5. This implies modified interfaces to the network environment for in the areas of charging and QoS support, and a modified authentication mechanism and modifications in the SIP signaling.

In this release of the PoC specification only interfaces between the network and the UE will be specified. Therefore Im, Is, and It are the only interfaces to be specified in this release of the PoC specification.

Nevertheless all core internal interfaces as depicted in the architectural picture above shall be based on IMS. All relevant core internal interfaces (including, but not limited to ISC) will be specified in subsequent releases of the PoC specifications.

**NOTE:** When PoC service is offered through a mobile network according to 3GPP release 5, and subsequent IMS releases, IMS signaling as defined by 3GPP shall be complied to.

## 5 Description of functional entities

### 5.1 User Equipment (UE)

The UE is the terminal equipment containing the PoC application client software.

### 5.2 IMS Core

The IMS core includes a number of SIP proxies and SIP registrars according to [6] and [13]. The first point of contact for UE is one of the proxies in the IMS core that is configured on the UE as the outbound proxy. In the IMS architecture, the outbound proxy is known as the P-CSCF. The IMS Core performs the following functions:

- Routes the SIP signaling between the UE and the PoC Server;
- Terminates the SIP compression from the terminal;
- Authentication and authorization;
- Maintains the registration state and the SIP session state;
- Reporting to the charging system.

The UE shall send all its SIP messages to the IP address of the outbound proxy after resolving the SIP URI of the outbound proxy to an IP address.

### 5.3 Group and List Management Server (GLMS)

PoC users use the GLMS to manage groups, contact lists and access lists.

A contact list is a kind of address book that may be used by PoC users to establish an instant talk session with other PoC users or PoC Groups. A user may have one or several contact lists including identities of other PoC users or PoC Groups. Contact list management includes operations to allow the UE to store and retrieve the contact lists located in the GLMS.

The end users can define PoC groups. The end user may select one group from the list to initiate an instant group talk session or a chat group talk session depending on the type of the group.

An access list is used by the end user as a means of controlling who is allowed to initiate instant talk sessions to the end user. An access list contains end user defined identities of other PoC end users or groups. The end user may have one blocked identities list and one granted identities list.

### 5.4 PoC Server

The PoC Server contains the PoC service. The PoC Server performs the following functions:

- End-point for SIP signaling;
- End-point for RTP and RTCP signaling
- Provides SIP session handling
- Provides policy control for access to groups
- Provides group session handling.
- Provides access control
- Provides do not disturb functionality.
- Provides the floor control functionality;

- Provides the Talker identification
- Provides the Participants information
- Provides the Quality feedback
- Provides the Charging reports
- Provides the Media distribution.

## 5.5 Presence Server

The presence server is not standardized in PoC release 1.0.

The Presence Server shall manage presence information that is uploaded by the Presence User/Network/External agents, and is responsible for combining the presence-related information for a certain presentity from the information it receives from multiple sources into a single presence document.

# 6 Description of the interfaces

## 6.1 Interface Is: UE – IMS Core

The Is interface supports the communication between the UE and the IMS Core.. This communication includes the SIP procedures defined in [2] to support the PoC features defined in [1].

The protocol for the Is interface is SIP (as defined by [13], other relevant IETF RFCs and necessary additions from [8]). The SIP signaling across this interface is subject to SIP signaling compression defined in [2]. The Is signaling shall be transported on UDP.

NOTE1: TCP transport for the Is interfaces is not supported in PoC release 1.0.

NOTE2: Presence service interactions between the UE and the IMS core will be specified in a future PoC release.

## 6.2 Interface If: IMS Core – PoC Server

The protocols over If interface support the communication between the IMS core and the PoC Server for session control.

NOTE: The If interface is not specified in the PoC release 1.0.

## 6.3 Interface It: UE – PoC Server

The protocols over It interface support the transport of talk bursts, floor control and link quality messages between the UE and the PoC Server.

The protocols for the It interface are RTP and RTCP [11] with details described in [3].

## 6.4 Interface Im: UE – GLMS

The protocols over Im interface support the communication between the UE and the Group and List Management Server (GLMS) for the purpose of managing the groups, contact lists and access lists and Do-not-Disturb indication. The HTTP/XML protocols shall be used as defined in [5].

The group and list management operations are detailed in [5].

## 6.5 Interface Ik: PoC Server – GLMS

The protocols over Ik interface support the communication between the PoC Server and the GLMS; enabling the PoC Server to retrieve the groups and access lists from the GLMS.

NOTE: The Ik interface is not specified in the PoC release 1.0.

## 6.6 Interface Ips: IMS core – Presence Server

The protocols over Ips interface enable the uploading of the registration status from the IMS core to the Presence Server and the dissemination of the presence information between the presence server and the UE.

NOTE: The Ips interface is not specified in the PoC release 1.0.

## 6.7 Interface Ipl: Presence Server – GLMS

The protocol over Ipl interface enables the uploading of the Do-not-Disturb status and the granted/blocked access lists from the GLMS to the Presence Server.

NOTE: The Ipl interface is not specified in the PoC release 1.0.

# 7 System concepts

## 7.1 Identification

### 7.1.1 Address of record (a.k.a. public user identity)

The PoC operator shall assign, to each user, an address-of-record (also known as public user identity) in the form of a SIP URI where the user part of the URI uniquely identifies the user, and this could be an alphanumeric string. The host part of the URI uniquely identifies a domain owned by the operator. The address-of-record shall comply with the specification of a SIP URI in [13].

The address-of-record is used for PoC only or for PoC and other SIP based services.

Examples of address-of-records are:

- sip:joe.doe@operator.net;
- sip:buss2.city@operator.net;
- sip:buss2.city@poc.operator.net.

### 7.1.2 Private user identity

The network operator shall assign to the user a private user identity for authentication purposes. The private user identity shall be hidden from the other users and cannot be used for addressing.

The private user identity shall take the form of an NAI, and shall have the form `username@realm` as specified in section 3 of RFC 2486 [22]. It is configured in the UE.

### 7.1.3 Group identities

The group identity used on the Is interface (between the UE and IMS core) for group talk, shall be generated by the GLMS and shall comply with the specification of a SIP URI in [13]. The GLMS shall generate the group identity according to reference [5].

### 7.1.4 Contact list identities

A contact list identity uses the generic address format defined in [13] and shall be used to address the contact lists in the GLMS. The GLMS shall generate a URI for each contact list as specified in reference [5].

## 7.2 Addressing

### 7.2.1 IP addresses

Each entity in the PoC system shall be assigned one or more IP addresses belonging to public or private IP realms.

IPv4 is mandatory.

### 7.2.2 Phone numbers

A PoC user may address another user by a phone number. The UE shall send the phone number to the IMS core in a TEL URL [23].

The phone number may use the international E.164 [21] format (prefixed with a '+' sign), or a local format using a local dialing plan and prefix. The IMS core shall interpret the phone number with a leading '+' to be an E.164 number.

Addressing by TEL URL for a PoC session requires that the PoC Server can resolve the TEL URL to a SIP URI, for instance by using DNS/ENUM or other local data base. A phone number in a local format shall be converted to the E.164 format before DNS/ENUM is used.

### 7.2.3 SIP URI

A PoC user may address another user by a SIP URI.

## 7.3 Routing principles

### 7.3.1 Registration

The PoC user shall register itself with the IMS core before using the PoC service.

The UE binds the public user identity and the UE IP address and port at registration with the IMS core. The UE shall include the public user identity as the address-of-record in the To header and the IP address as the host part of the SIP URI in the Contact header.

The UE shall include the media feature tag [19] indicating the PoC service in the Contact header. The IMS core may use this information for routing.

### 7.3.2 Session establishment

The INVITE request on the Is interface shall contain the Accept-Contact header [19] with a media feature tag indicating the PoC service. The IMS core shall be able to identify the request as PoC communication by inspecting the Accept-Contact header.

The Request-URI of the INVITE shall contain either the pre-configured ad-hoc identity (for instant personal talk and ad-hoc instant group) or a group identity (for instant group talk or chat group talk). The pre-configured ad-hoc identity is assigned according to reference [2].

Early session establishment is used for having session available for quick connection establishment with REFER. Early session establishment's INVITE does not have any referred party field and can be differentiated from this against other INVITEs.

The transient group identity is generated by the PoC server and distributed to the UE in the contact header.

From the initiating UE, the public user identity of the inviting user shall be included in the From header. On the signaling towards the invited user, the From header includes either the public user identity (instant personal talk, ad-hoc instant group) or the group identity (instant group talk or being added to a chat group).

### 7.3.3 User location

Users shall use the Address of record or phone number (as described in 7.2.2 and 7.2.3) of other users to address them. The IMS core is responsible for routing the requests to the users according to the bindings established through registrations.

## 7.4 Security

### 7.4.1 Threats

There are a number of SIP threats identified in [13]. The main problem is that it is simple to steal and use another user's identity unless the identity is properly authenticated. The main threats by unauthorized use of an identity are according to [13]:

- Registration Hijacking;
- Impersonating a Server;
- Tearing Down Sessions.

Another threat is fraud where a malicious user steals another user's identity to talk and let the other user pay for that.

### 7.4.2 Countermeasures

The best and simplest countermeasure to the threats above is to properly authenticate the SIP requests from a user. The Digest procedure defined in [13] shall be supported between the UE and IMS core for authenticating the user. The UE shall include any stored credentials in all relevant SIP requests to avoid unnecessary signaling. With the digest procedure, the IMS core should challenge the user if a relevant SIP request lacks credentials or the credentials are invalid. The details for the authentication procedure are defined in [2].

A user shall be authenticated with the GLMS by using the procedures defined in [5].

In the case of HTTP digest, then the same digest password should be used by the UE when communicating with GLMS or IMS core. The UE should not prompt the user for a digest password if the password is configured in the UE. Any configured password should be protected, for example, by encryption.

## 7.5 Floor control

Floor control includes the following actions:

- floor request;  
The action provides the capability for a participant in a talk session to ask for permission to talk.
- floor release;  
The action taken by a granted user to release their permission to talk.
- floor grant;  
An action from the network to inform requesting participant that the floor has been granted.
- floor idle indication;  
An action from the network to inform participants that the floor is idle.
- floor deny;  
An action from the network to inform the requesting participant that the floor request is denied.
- floor taken;  
An action from the network to inform all participants that the floor has been granted to the indicated user.
- floor revoke;  
The action from the network to remove the permission to talk from a user who has previously been granted the floor
- talk burst quality feedback.  
reports the amount of media received by the PoC server or by the UE.
- RTCP BYE;  
indicates that the sending party wants to terminate the ongoing media session in current communication context, without changing the SIP-session state

The talker arbitration performed through the use of RTCP.

The talker identification distributed through the use of RTCP

The talk burst quality feedback is distributed through the use of RTCP RR and SR

Floor control is described in detail in [3].

## 7.6 Codecs

PoC has the AMR speech codec as the mandatory speech codec for the GSM, GERAN and UTRAN radio access networks. The details describing how AMR is used in PoC is found in [3]. The codec rate depends heavily on the radio cell configuration and the latency for GSM/GERAN/UTRAN.

## 7.7 Signaling compression

The UE and the IMS core shall compress the SIP signaling according to [15], [16], [17] and [18] to reduce the transmission delays. Details how this is done are described in [2].

The grouping of messages (compartments) starts at registration and ends at de-registration.

A static SIP dictionary [17] shall be stored locally in the UE and the IMS core. A user specific dictionary [16] may be uploaded by the UE to the IMS core and stored there as part of the registration procedure. The user specific dictionary is used to increase the compression rate in particular for the initial SIP messages. The SIP URI shall include the signaling compression indication as described in [18].

## 7.8 User plane adaptation

The media throughput is affected by the conditions on the radio access link. The talk burst (user plane) bit rate shall be reduced in case the talk burst bit rate is higher than the available end-to-end bit rate. The available bit rate is indicated with RTCP. The talk bursts bit rate can be reduced if necessary by re-negotiation with SIP.

## 7.9 Charging

The charging models that should be supported are described in [1]. The PoC architecture shall be able to support the following offline charging mechanisms to support these charging models:

- The IMS core network nodes and the PoC Server shall be able to report to the charging system upon reception of various SIP methods since charging relevant information is contained in these messages;
- The PoC Server shall be able to report a talk burst end event to the charging system. The message sent to the charging system for the talk burst should include any relevant SIP and SDP parameters for the ongoing session, the numbers of bytes sent, the number of packets sent and the duration of the talk burst.
- The PoC Server shall be able to report to the charging system the number of packets and bytes a UE has received in a talk burst. The UE reports these parameters in a receiver report message;
- The delivery of the receiver reports and the talk burst end are not reliable. The PoC Server shall be able to report to the charging system after a timer expiry in case both of these messages are lost.

The receiver report cannot be trusted. In case there is a long-term difference between the number of bytes sent from PoC server and the information received in the receiver report, this is an indication of fraud.

The bearer charging is also used in parallel. The correlation between the bearer charging and service charging shall be done in the charging system.

A deployed network may implement all, some or none of the charging models described in this sub-clause.

## 7.10 Roaming

The PoC user may roam in a visited network. GPRS roaming can then be used which means that a home GGSN and the home IMS core are always used.

## 7.11 Presence

For further study

## 7.12 Do-not-Disturb

The user can enable the Do-not-Disturb (DnD) feature in the UE. The network or the UE with the enabled DnD will then reject all incoming sessions. The inviting user does then recognize the response as a busy indication.

The user updates the DnD setting in GLMS. This is described in detail in [5].

# 8 High level procedures

The clause gives examples on PoC signaling procedures. Details are described in the Signaling flows [2]

## 8.1 PDP Contexts

A PDP Context with the interactive traffic class with the highest priority should be used for the SIP, DNS and HTTP/XML signaling. The signaling PDP Context is a Primary PDP Context that shall be activated before the SIP registration. This PDP Context should be activated at least as long as the SIP registration lasts.

In case the radio access network does not support the streaming class or the streaming class usage is subject to local policy, a PDP Context with the interactive traffic class with the highest priority should be used for media. The Primary PDP Context may be used for the media. As an alternative a Secondary PDP Context with the interactive traffic class may be used for the media instead of the Primary PDP Context.

In case the radio access network supports the streaming traffic class and the local policy allows its usage PDP Context with the streaming traffic class should be used for the media to provide high quality voice. The PDP Context for the streaming traffic class is a Secondary PDP Context that is activated in parallel with the establishment of Instant Personal Talk or Group Talk.

## 8.2 DNS name server discovery

The UE obtains IP addresses of the DNS name servers as part of the PDP context activation.

The IPv4 addresses for the DNS name servers (primary and secondary) are included in the Protocol Configuration Options (PCO) [7]. The PCO format uses the IPCP [9] field that is defined in [10].

## 8.3 DNS procedures

### 8.3.1 SIP URI resolution

The address of the outbound proxy in the IMS core is configured into the UE in the form of a SIP URI. The URI should include the transport protocol (UDP) and the port number the proxy is listening to and an indication that compression should be used (comp=sigcomp).

In order to contact the outbound proxy, the UE performs a DNS query on the host part of the SIP URI of the proxy following the rules described in [14]. Note, however, that if the host part is an IP address rather than an FQDN the UE does not need to consult the DNS.

Note that if the SIP URI of the outbound proxy that is configured in the UE follows the format recommended above (transport protocol and port number explicitly given), following the procedures in [14], the UE performs an A query. If the recommended format is not followed, the UE might need to perform NAPTR or/and SRV queries.

Example of the recommended format for the SIP URI configured in the UE pointing to the outbound proxy:

```
sip:outbound-proxy.my-operator.com:5060;transport=udp;comp=sigcomp
```

### 8.3.2 HTTP URL resolution

The UE resolves a HTTP URL according to [12] or to best current practice.

**NOTE:** The Im request is sent to a proxy identified by a configured address in case a WAP or HTTP proxy is used. In that case the proxy resolves the HTTP URL according to [12] or to best current practice.

## 8.4 Early Session dialog establishment

The early session provides the instant personal communication for end-user. The early session enables the UE to communicate the IP addresses and ports, which are used for sending the RTP/RTCP packets.

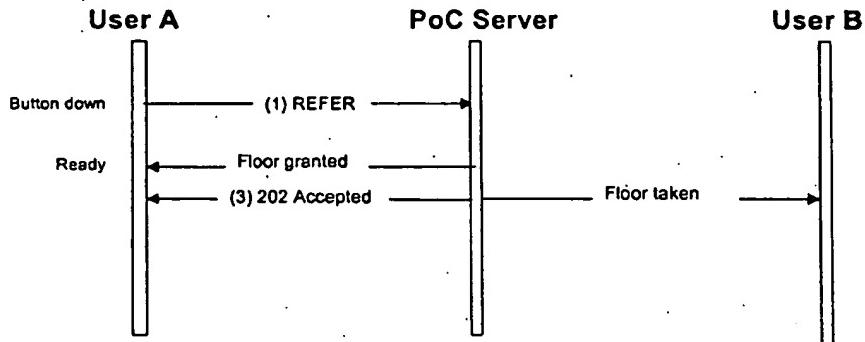
The early session is used for service negotiation purposes between the end user and his/her home PoC application server.

The early session is established for service negotiation purposes right after the initial registration, or the UE may request the early session for instant personal communication at any later time point (e.g. when the end-user activates the instant personal communication from the UE).

## 8.5 Instant Personal Talk

In an Instant Personal Talk, user A establishes an PoC session with user B (via the PoC Server). Connection setup may be done either with early session procedure and SIP REFER or with SIP INVITE.

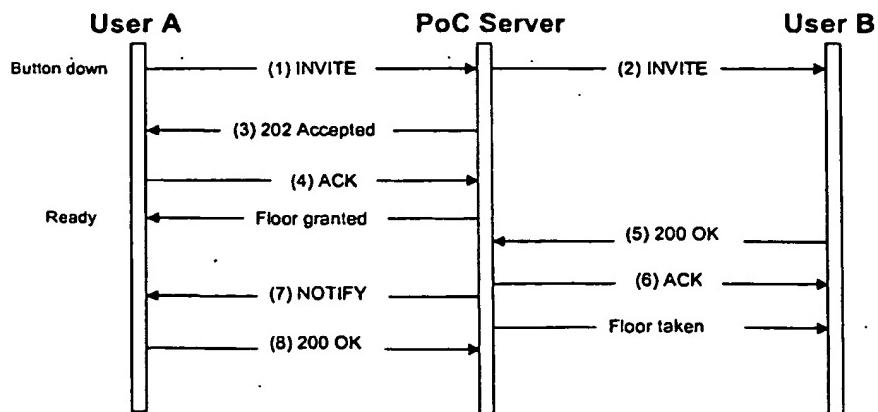
In early session case the User A and User B establish early sessions after registration towards their PoC servers and during the connection establishment phase the session is established with using SIP REFER. If user A's terminal supports early session and auto answer is defined for user B Figure 2's signalling sequence is applied. NOTIFY message is sent to user A after the first talk burst.



**Figure 2: Early session and auto answer procedure**

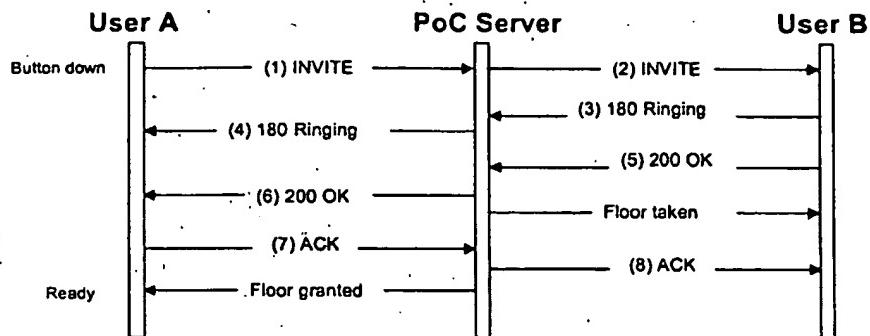
User A sends a SIP INVITE to the PoC Server including user B's address in a body whose type is application/vnd.siemensericssonmotorola.PoC. When the PoC Server receives this INVITE, it performs an authorization decision based on the Do-not-Disturb setting in the GMLS. If the invitation is authorized, the PoC Server inspects the application/vnd.siemensericssonmotorola.PoC body to discover user B's address and, depending on the network configuration, it initiates early media procedures (optional) or late media procedures.

If the PoC Server is configured to use the optional early media procedures, it will, as shown in Figure 3, answer the INVITE with a 202 (Accepted) response. This response, together with the "floor grant" message from the talker arbitration process, informs user A that user B has not been reached yet, but that the PoC Server is already prepared to receive media. The PoC Server will buffer all the media received from user A until it can be delivered to user B. When user B is finally contacted, the PoC Server informs user A about this using a NOTIFY request.



**Figure 3: Early media and auto answer procedure**

If the PoC Server is configured to use the late media procedures, it will, as shown in [Figure 3](#) simply relay the responses received from user B to user A.



**Figure 4: Late Media and Manual answer procedure**

Once the session has been established, either user A or user B can terminate it at any time. In case of inactivity, the PoC Server can also terminate the session (no talk bursts within a configured time period).

## 8.6 Ad-hoc Instant Group Talk

In an Ad-hoc Instant Group Talk, user A establishes a PoC session with a set of users (via the PoC Server).

For the early media procedures and the late media procedures are similar to those described in figures 3 and 4. The difference is that in an Ad-hoc Instant Group Talk, the application/vnd-poc.refer-to body of the initial INVITE from user A will contain more than one address. Therefore, the PoC Server will send more than one INVITEs as a result. However, the fact that there are multiple INVITEs triggered by user A's initial INVITE is hidden from user A. The PoC Server will send to user A a single NOTIFY in the early media case and a single 180 (Ringing) response in the manual answer case. When using the optional early media procedures for ad-hoc groups, the PoC server begins to play the media when the first invited party accepts.

For the early session case procedures are similar to those described in figure 3. The difference is that in the REFER from user A will contain more than one address. PoC server will initiate several session setups towards B parties. The setup will be done with auto or manual answering mode depending on the Bs configuration.

Note that, according to the description above, an instant personal talk (sub-clause 8.4) is only a special case of an Ad-hoc Instant Group Talk. Therefore, in all cases the user portion of the Request-URI of the initial INVITE will be set to the pre-configured ad-hoc string. The PoC Server will return in the Contact header of the final response for the INVITE a unique identifier for the Instant Personal or Instant Group Talk.

## 8.7 Instant Group Talk

The signaling messages exchanged to establish an Instant Group Talk are the same as for an Ad-hoc Instant Group Talk. The only difference is that the Instant Group Talk has an already existing identifier while the identifier for the Ad-hoc Instant Group Talk was created at establishment time. Therefore, the Request-URI of the initial INVITE will be set to the identifier of the Instant Group Talk, as opposed to the Ad-hoc case where the "ad-hoc" convention for the user part is used.

## 8.8 Chat Group Talk

In a Chat Group Talk, a user joins an existing chat group. Existing, in this context, means that a unique identifier has been previously allocated for the chat group. Thus, both Chat Group Talks and Instant Group Talks have previously existing identifiers. The difference is that in an Instant Group Talk, when the first user joins the group, all the members of that Instant Group Talk are invited by the PoC Server. In a Chat Group Talk, when a user joins the group, no other user is implicitly invited.

## 8.9 Instant Personal Alert

An Instant Personal Alert is a request from user A to user B for user B to establishes an Instant Personal Talk with user A. Instant Personal Alerts are similar to a call back service in the world of traditional telephony. They are implemented using the SIP MESSAGE method. User A is informed of the delivery result of the instant personal alert.

---

## ANNEX A (informative): Known 3GPP/OMA divergence

PoC industry specification phase 1 is based on 3GPP and IETF standards where feasible. In order to overcome limitations of the existing technology, the PoC phase 1 specifications contain some deviations from the 3GPP standards. Careful evaluation of these deviations is required in subsequent releases of Push-to-talk specification. These divergences included in the industry specification do not imply requirements on the 3GPP/OMA standards but the OMA standards work should follow and be aligned with the 3GPP IMS specifications as much as possible.

This annex captures the known divergences from the 3GPP specification.

- **IP version**  
The PoC phase 1 specification supports IPv4 only
- **Authentication and integrity protection**  
The PoC phase 1 specification supports HTTP digest in place of the IMS-AKA specified by 3GPP.
- **Policy control**  
The PoC phase 1 specification does not provide for the support of the IMS headers related to the policy control (Go interface).
- **CSCF discovery**  
The PoC phase 1 specifications describe the parameters which are required to be configured – this includes the contact address of the SIP proxy which the terminal will communicate with.
- **Provisioning of private user ID**  
The PoC phase 1 specifications describes the parameters which are required to be configured – this includes the private user identifier. In 3GPP this is located on the ISIM or derived from the USIM.
- **Floor control**  
The PoC phase 1 specifications describes floor control. There is no suitable floor control standardized within the industry.
- **Group List Management**  
The PoC phase 1 specifications describes a group list management protocol. The group list management work in 3GPP is still ongoing.
- **PoC header extension**  
The PoC phase 1 specifications describes header extensions for identifying that the signaling is related to the PoC service.
- **Initial, Re- & De-registration procedures**  
The PoC phase 1 specifications do not use all the 3GPP specified SIP header fields and parameters and additionally the PoC phase 1 specifications use PoC specific header fields and parameters.
- **Reject and Error Codes in Sessions**  
The PoC phase 1 specifications have specific usage of reject and error codes.
- **Conferencing**  
The overall POC functionality has similarities with currently ongoing 3GPP Rel6 conferencing work. Further alignments of these two activities are needed.
- **Group Identities**  
The PoC phase 1 specification defines a mechanism for creating group identities. 3GPP has not yet completed the work on group management including handling of group identities and the Ut reference point.
- **Implicit subscription to refer state event**  
The PoC phase 1 specifications establish with SIP INVITE method an implicit subscription which is not specified in 3GPP nor IETF.

---

## Annex B: Change History (Informative)

Date	Subject/Comment	Old	New
2003-08-25	Agreed by Ericsson, Motorola, Nokia, Siemens		v.1.1.0
2003-10-06	"Confidential" and "proprietary" notes removed.	v1.1.0	v.1.1.1